6.456 Lab 1

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Pre-Lab

1. Find the resistor values for first-order filters for the following frequencies and capacitor values:
fc = 1/(2*pi*R*C)
R = 1/(2*pi*C*fc)
1a) fc = 2000 Hz, C = 0.1 uF, R= 796 Ohms
1b) fc = 3000 Hz, C = 0.1 uF, R= 531 Ohms

2. Define the following terms that apply to data collection using a DAQ (data acquisition system). For reference, here is a link to the manual of the DAQ we will use in this class: https://www.mccdaq.com/pdfs/manuals/USB-1608FS-Plus.pdf
Sample rate: The number of samples per second
Bit depth:16-bit - the accuracy with which the analog signal is converted to digital
Gain:A scale factor for scaling the signal into the full range of the DAQ. Select the input range that is most appropriate for the signal.
Number of channels:8 - number of input signals that can be measured simultaneously
Number of samples:up to 32,768 samples - the memory size of the data buffer; divided by the sample rate gives the longest duration of measurement you can take at one time

Part A: Tools of the trade

Station 1: Analog Filtering

1.1.1: Provide the amplitude v. frequency for the table above and plot. Compare the result to the expected first-order bandpass filter, and describe any differences

Answer: Scope 1 should present us with the actual signal input voltage to the system. In scope 2 we should read the same signal with a low pass filter applied to it with a cutoff frequency of 5.305 kHz and a high pass filter with cutoff frequency 796 Hz. The main difference in the observed result to the expected first-order bandpass filter comes in Scope Position 3 when R1 = 300 Ohms and R2 = 2 kOhms. The observed results in amplitude are higher than the input value into the circuit. The observed amplitudes are far greater than the generated signal, which is not possible unless there is an amplifier in the circuit. Due to having no amplifier in the circuit, we note that there is some error in that portion of the lab. Photos of the results are attached, where the colors are described as follows: Channel 1 (Yellow), Channel 2 (Pink), Channel 3 (Blue).

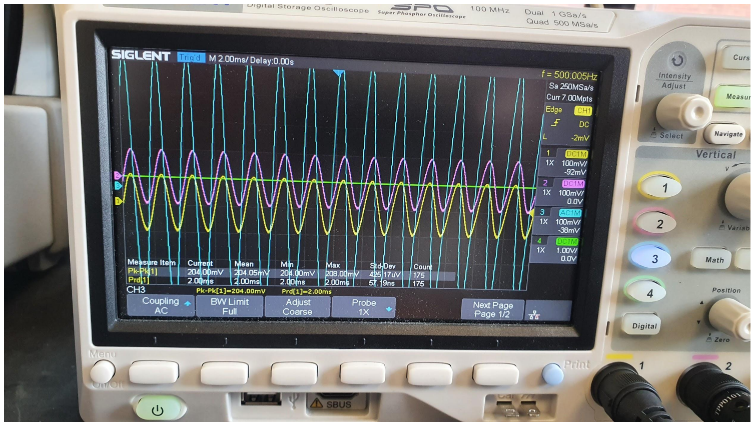


Fig 1. Result from Lab, Station 1 Part 1 at 500 Hz.

Fig 2. Result from Lab, Station 1 Part 1 at 1500 Hz.

Fig 3. Result from Lab, Station 1 Part 1 at 2500 Hz.

Fig 4. Result from Lab, Station 1 Part 1 at 3500 Hz.

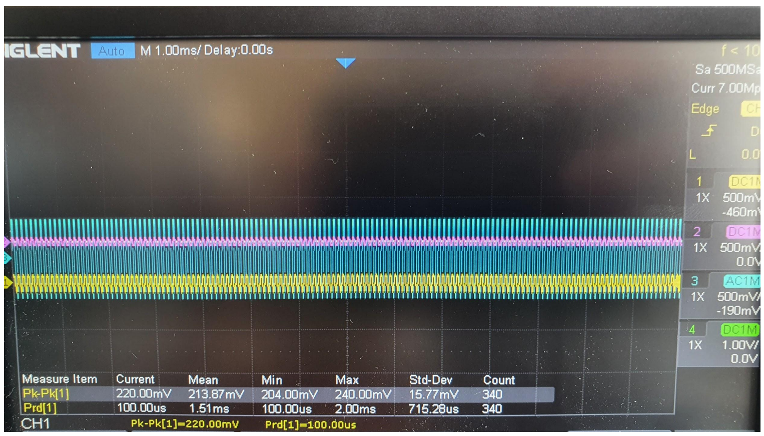


Fig 5. Result from Lab, Station 1 Part 1 at 10000 Hz.

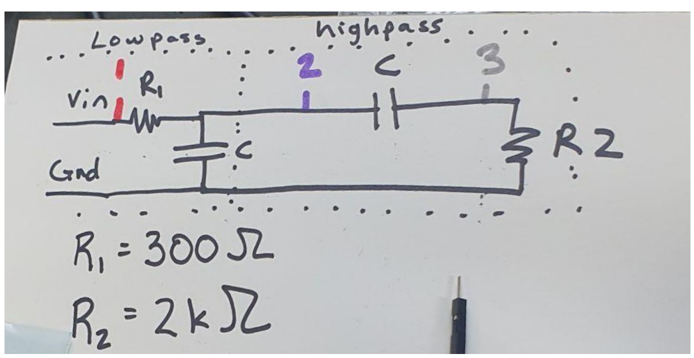
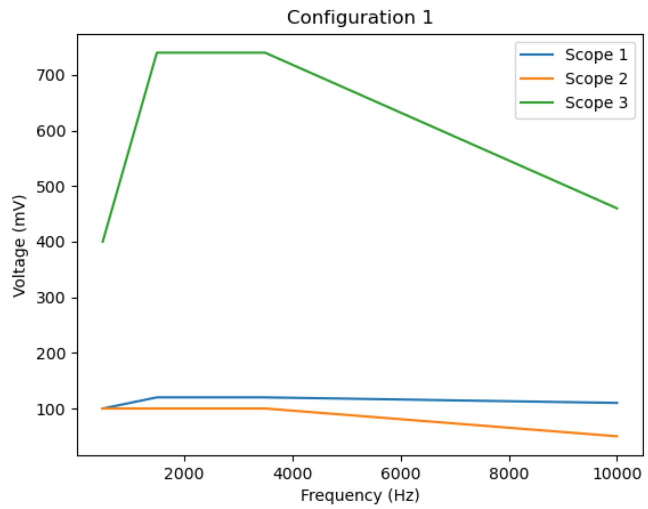


Fig 6. Analog filter configuration and oscilloscope probe locations



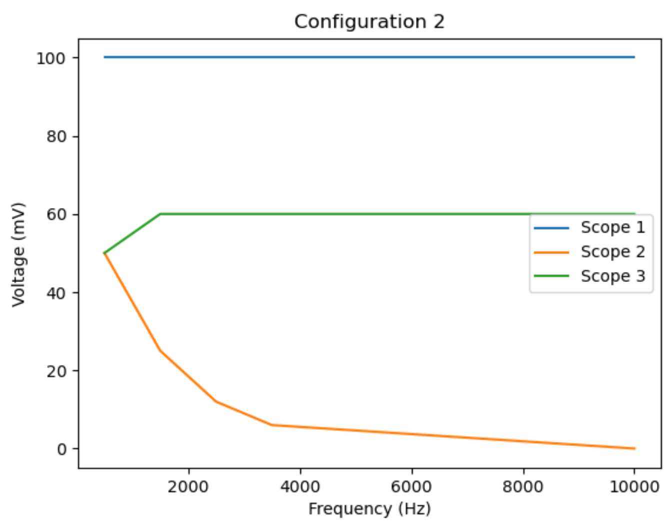


Fig 7. Voltage amplitude output in the 3 scopes for both analog filter configurations in (mV) as a function of signal input frequency

Table 1 - Original resistor configuration

Frequency	Scope 1 Amplitude	Scope 2 Amplitude	Scope 3 Amplitude
500 Hz	100 mV	100 mV	400 mV
1500 Hz	100 mV	100 mV	740 mV
2500 Hz	100 mV	100 mV	740 mV
3500 Hz	100 mV	100 mV	740 mV
10000 Hz	100 mV	50 mV	460 mV

Table 2 - Switched resistor configuration

Frequency	Scope 1 Amplitude	Scope 2 Amplitude	Scope 3 Amplitude
500 Hz	100 mV	50 mV	50 mV
1500 Hz	100 mV	25 mV	60 mV
2500 Hz	100 mV	12 mV	60 mV
3500 Hz	100 mV	6 mV	60 mV
10000 Hz	100 mV	~ 0 mV	60 mV

1.1.2: What happened when you swapped the resistors in the two filters?
Answer: We moved from a lowpass filter with f_c of 5.305 kHz and highpass filter with f_c of 796 Hz to a lowpass filter with f_c of 795 Hz and a highpass filter with f_c of 5.305 kHz -- essentially swapping the frequencies of lowpass and highpass filters. Since no frequencies exist that fall within the allowed frequencies of both filters, all frequencies were attenuated.

1.1.3: Does the filter order matter?
Answer: Filter order does not matter. Whether the lowpass or highpass filter comes first makes no difference.

Station 2: Hydrophone

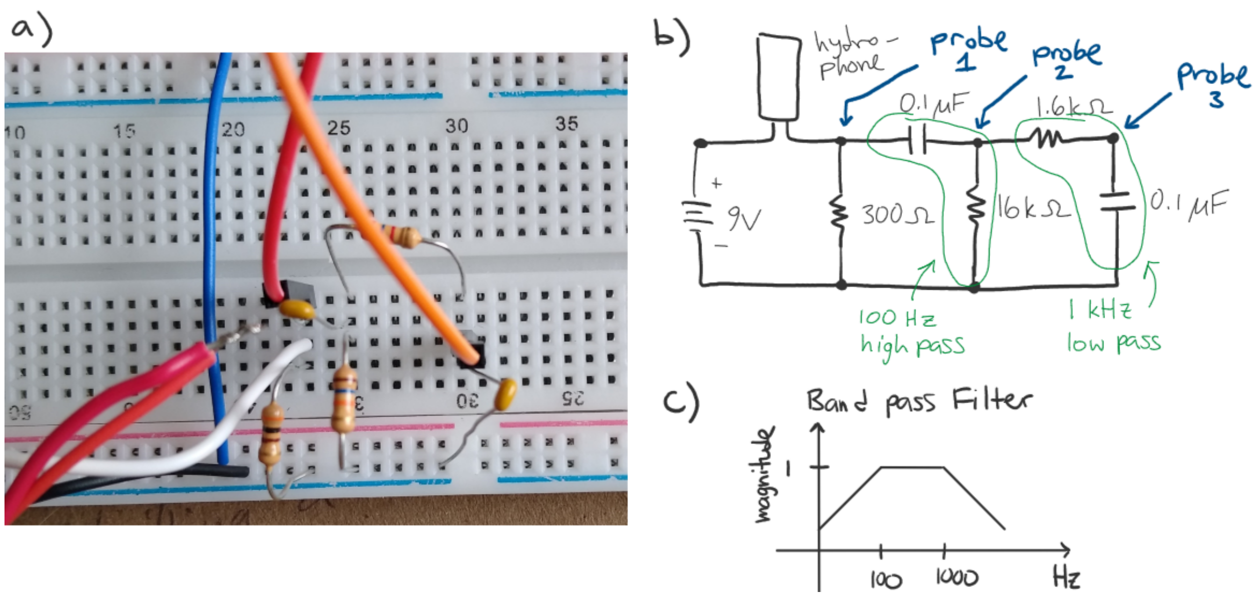


Figure 8: a) Photo of the circuit; b) Circuit schematic; c) Notional filter response magnitude plot

1.2.1 Why would we include a voltage divider in the circuit?
Answer: To bias the circuit output at a value between 0 and 9 V so that when the hydrophone sensed sound there would be room for the output to oscillate without going negative.

1.2.2 Why would we include a highpass filter in the circuit?
Answer: To filter out the DC bias signal from the voltage divider, and to eliminate unwanted low frequency noise (e.g. 60 Hz noise)

1.2.3 Why would we include a lowpass filter in the circuit?
Answer: To limit the highest frequency that is sensed because often sound is composed of the fundamental tone and higher frequency harmonics but maybe we only wanted to observe the fundamental.

1.2.4 What are the cutoffs for the highpass and lowpass filters in the circuit? What dictates these frequencies?
Answer: The lowpass cutoff was 100 Hz and highpass cutoff was 1 kHz. The cutoff frequency is dictated by the choice of resistor and capacitor in the filter.

1.2.5 What are examples of high frequency and low frequency signals we may want to exclude?
Answer: Examples of unwanted low frequency signals are DC bias and the 60 Hz buzz from power lines. An example of high frequency is the harmonics of the tone we are observing.

1.2.6 What is the effect of the filter on the oscillation?
See photo and description below:

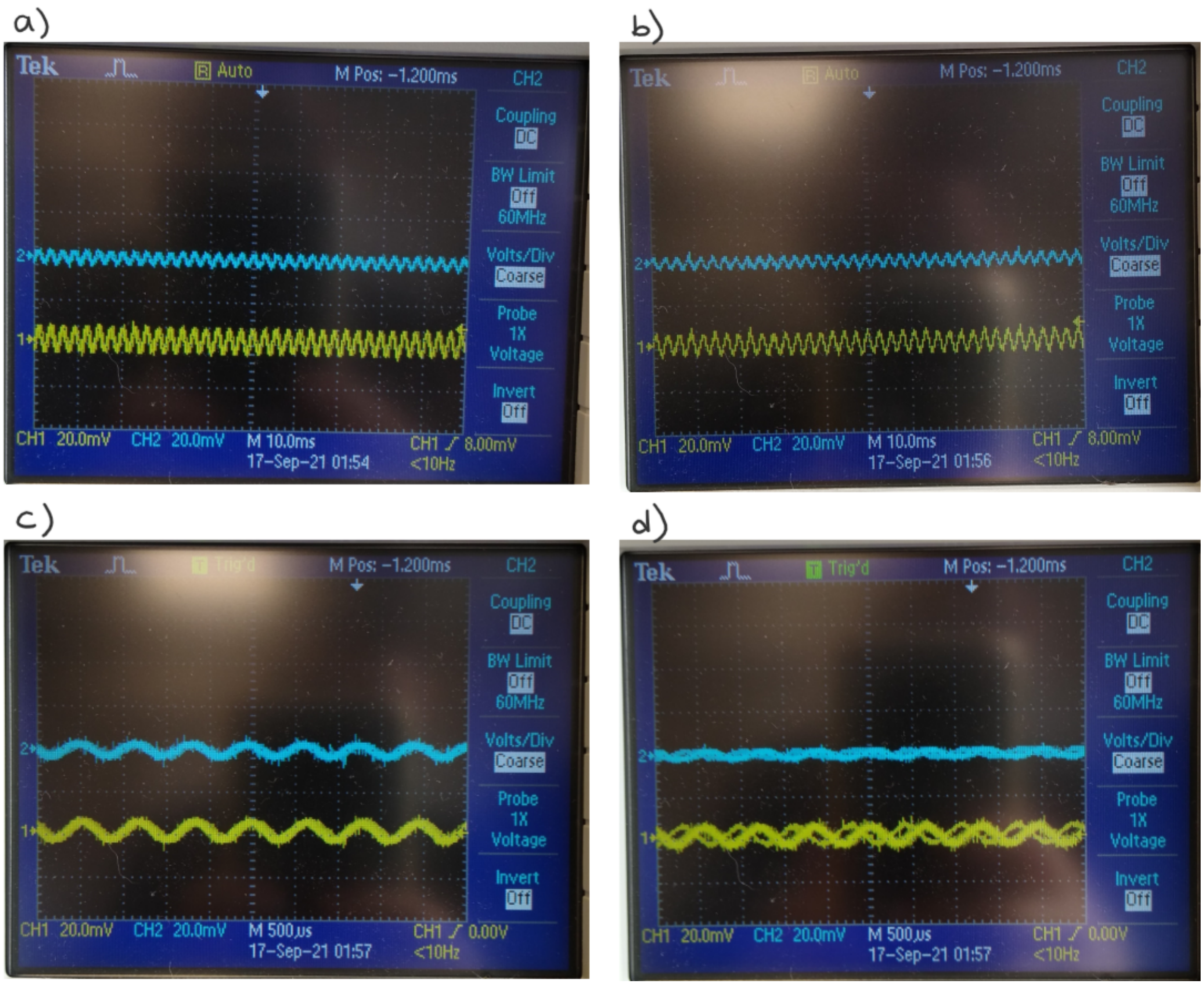


Figure 9: Oscilloscope Outputs,CH1 (yellow) is unfiltered, CH2 (blue) is filtered

A. Input frequency 366 Hz. Ch 1: raw input (AC-coupled); Ch 2: after high pass filter.
B. Input frequency 366 Hz. Ch 1: raw input; Ch 2: after low pass filter. Demonstrating that signal amplitude does not change after filtering because this signal is in the passband.
C. Input frequency 1535 Hz. Ch 1: raw input; Ch 2 after high pass filter.
D. Input frequency 1535 Hz. Ch 1: raw input; Ch 2 after low pass filter. Demonstrating that signal amplitude is decreased after the lowpass filter because this frequency is outside the passband.

1.2.7 What is the impact of the lower-frequency bandpass? Compare to the previous filter.
Answer: Moving from a (100 - 1000 Hz) bandpass to a (100 - 200) Hz bandpass adds additional attenuation to frequencies greater than 200 Hz. For example, a signal at 400 Hz would come through clearly in the first filter, but be greatly attenuated in the second. Frequencies below 200 Hz would be unaffected. And, as before, frequencies under 100 Hz would also be attenuated.

Station 3: DAQ and Anti-Aliasing

1.3.1 Plot the spectrogram for all the data.

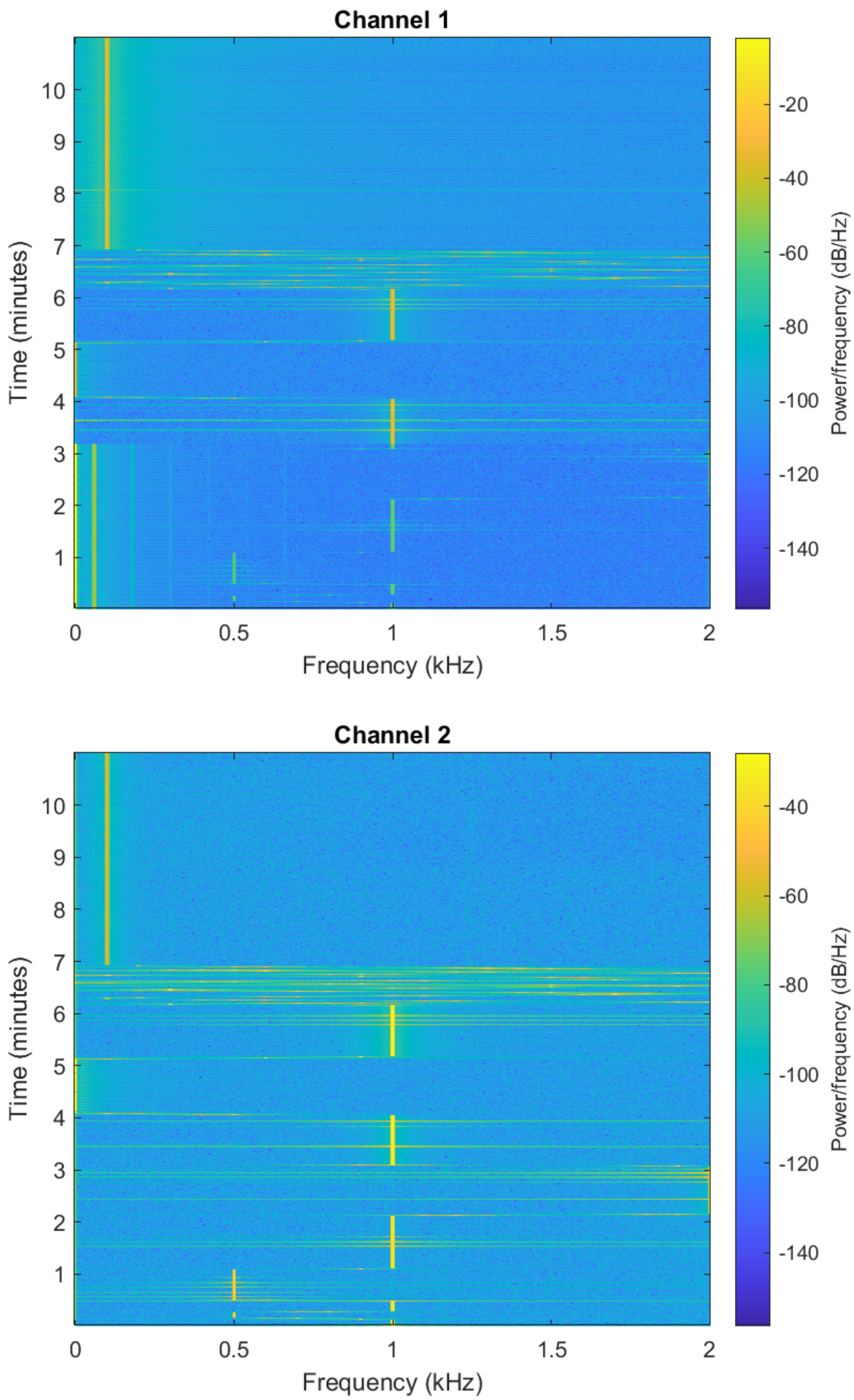


Figure 10: Channel 1 and 2 Spectrograms for DAQ

1.3.2 Plot the FFT and spectrogram. Observe the effects of aliasing below Nyquist frequency. What is the effect of aliasing on the data? How could aliasing change what you thought you were observing if you didn't know it was happening?

Aliasing causes frequencies at $f_c > f_s/2$ to appear identical to an aliased frequency $f_a = |f_s - Nf_c|$, where N is an integer chosen such that $0 < f_a < f_s/2$. For example, in the plots below, we can see that 4 kHz aliases to 0 Hz, and 5 kHz aliases to 1 kHz, for a 4 kHz sampling rate. If we were looking for a signal with expected frequency f_c, and there are other signals in the environment with spectrums that alias to f_c, we might have difficulty separating the two signals if we haven't applied a proper antialiasing filter on the front end.

Note: We had a large DC offset on channel 1 during the first two minutes, shown below. This explains why we have a large component at center frequency in the first four plots. We made an attempt to remove the offset during post-processing, but some low frequency components remain. However, spikes at the correct frequency offsets when the plots are compared.

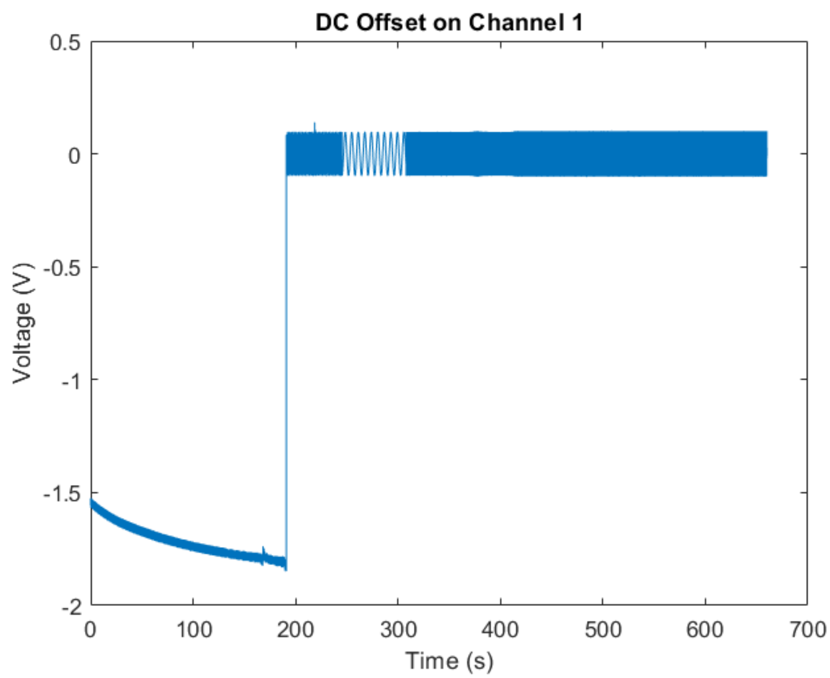


Figure 11: DC Offset on Channel 1

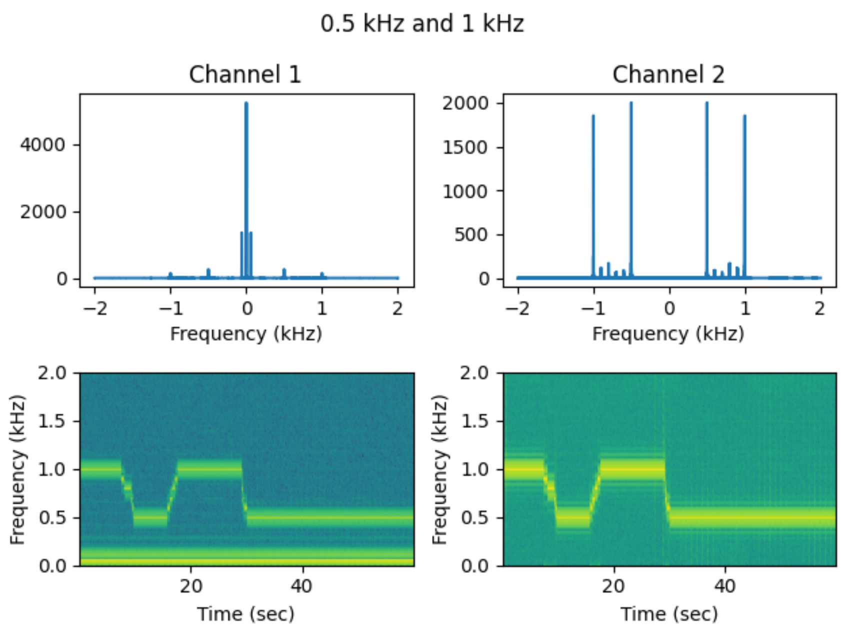


Figure 12: FFT and Spectrogram for 0.5 kHz and 1 kHz (Note DC offset)

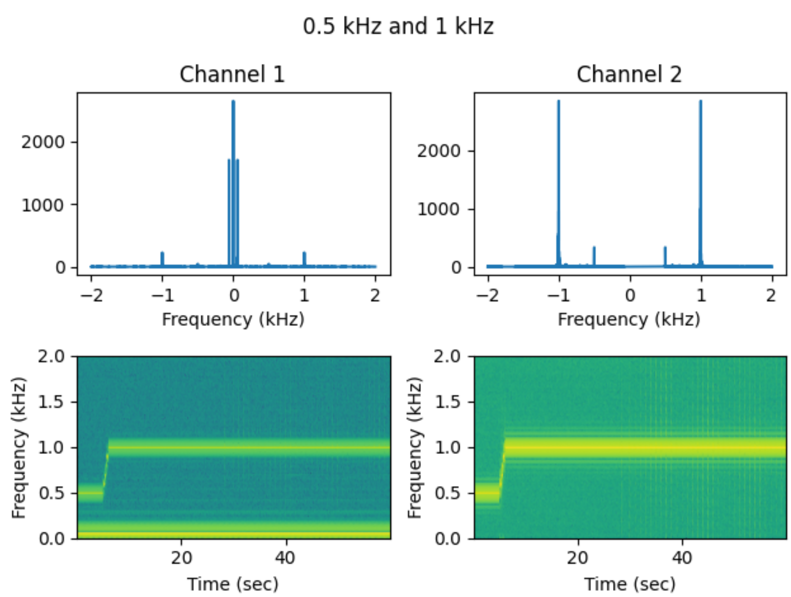


Figure 13: FFT and Spectrogram for 0.5 kHz and 1 kHz

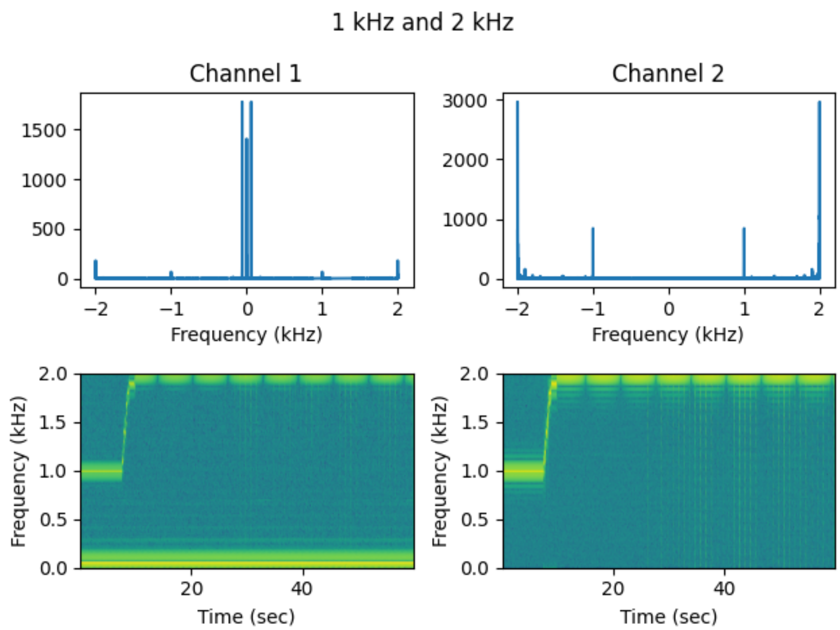


Figure 14: FFT and Spectrogram for 1 kHz and 2 kHz

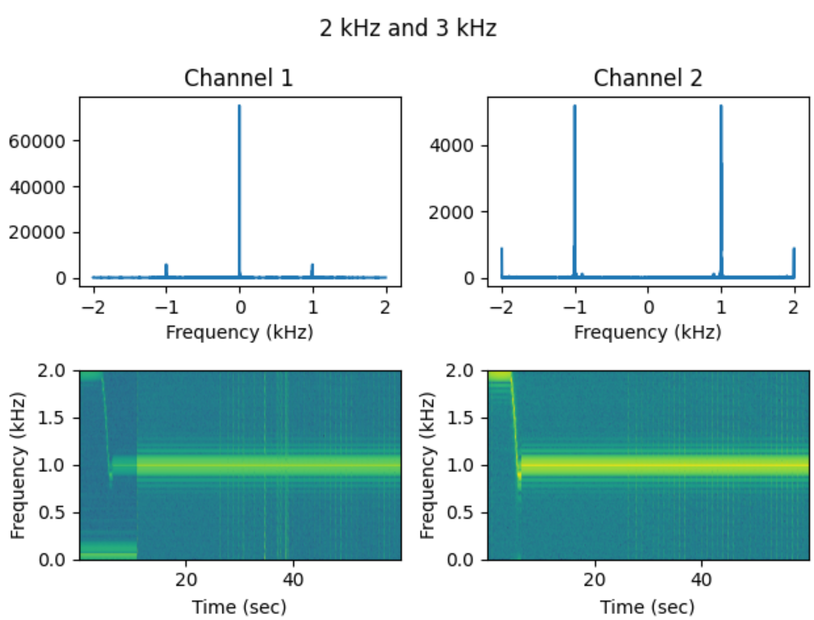


Figure 15: FFT and Spectrogram for 2 kHz and 3 kHz

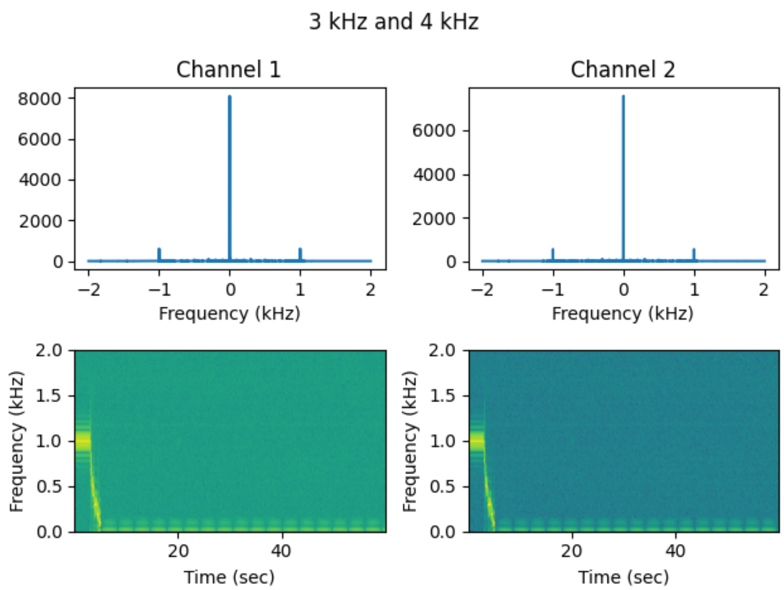


Figure 16: FFT and Spectrogram for 3 kHz and 4 kHz

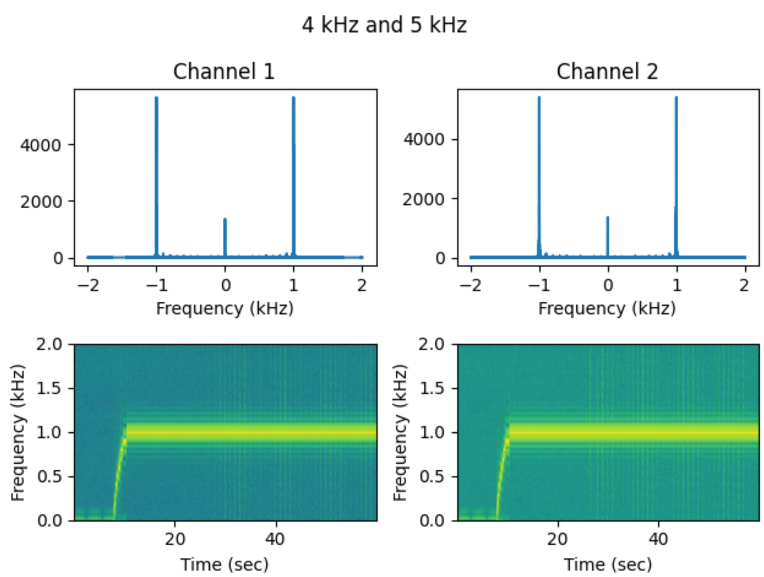


Figure 17: FFT and Spectrogram for 4 kHz and 5 kHz

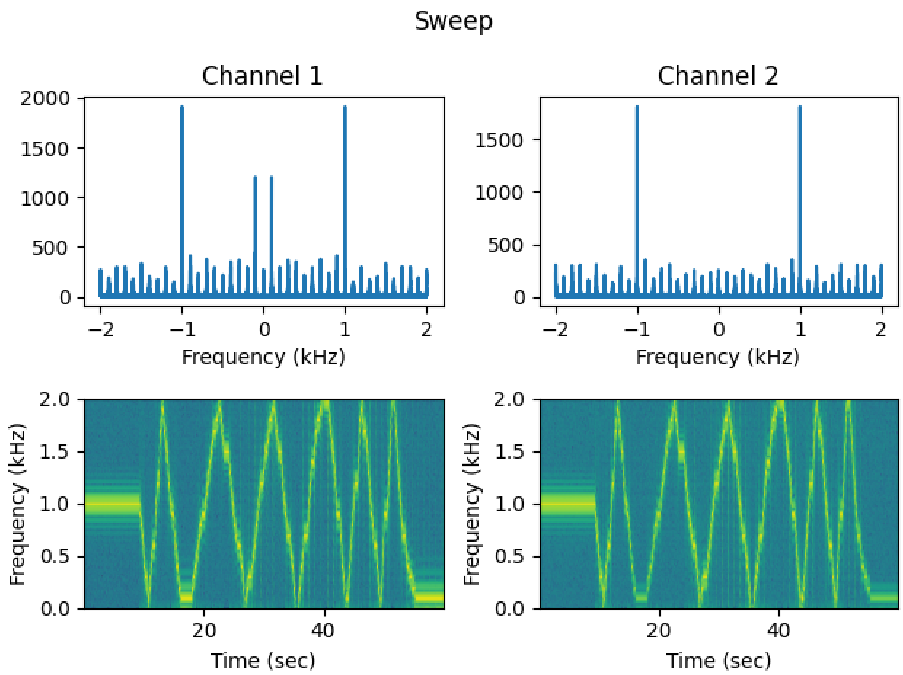


Figure 18: FFT and Spectrogram for Frequency Sweep

1.3.3 Observe the effects of aliasing with and without the anti-aliasing filter. What do you see?
Answer: Sadly, the anti-aliasing filter wasn't present as only a high-pass filter was applied on Channel 2. We do see aliasing from 3kHz signals and upwards, and also in the Sweep spectrogram. Moreover, we do see reduction of low frequency content on Channel 2 as a result of the high-pass filter that is being applied to the original signal shown in Channel 1. We can see the effects of the filter here, where the low frequencies are greatly attenuated in the figure below.

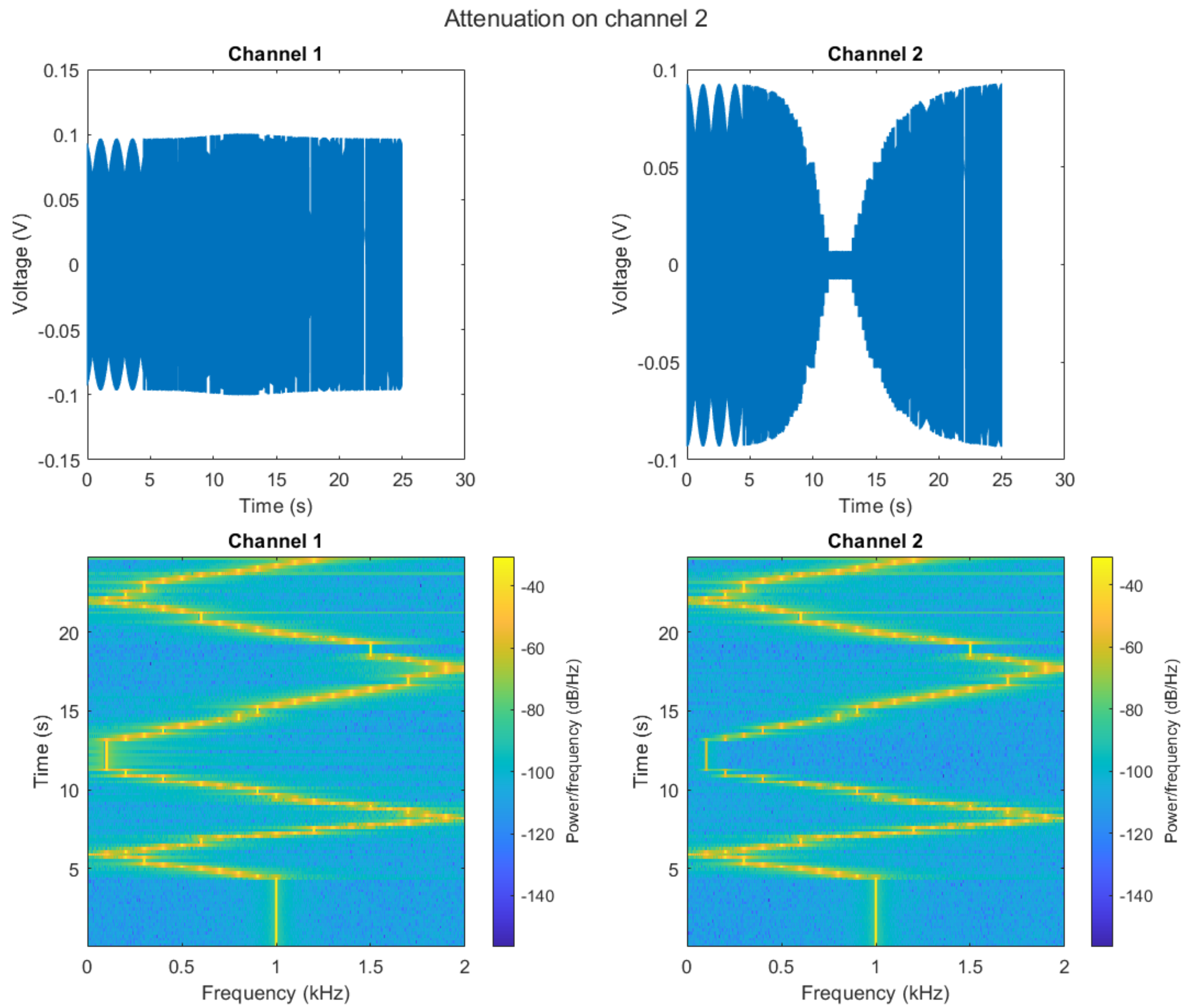


Figure 19: Effects of Attenuation on Channel 2

1.3.4 For the part of the data where the tone was fixed at 3.5 kHz, how would you process the data if you knew your signal was between (2 - 4.5) kHz to eliminate the effects of aliasing?
Answer: We would apply a bandpass filter between (2 - 4.5) kHz, as we know the signal of interest is within that range. Unfortunately, the bandwidth we must observe is greater than $f_s/2$, so aliasing cannot be completely eliminated.

Part 2: Array Data Collection: Not Yet Due

Photographs from Data Collection

Figures []: Testing Series

Array Data Collection: Uniform Array

Source	Position	Start Time	End Time
Ambient/Baseline	N/A	1429	1431
4 kHz	Fwd Endfire	1433	1435
4 kHz	045 Relative	1437	1439
4 kHz	Broadside	1440	1442
Noise 6 kHz	045 Relative	1454	1456
Noise 4 kHz w/	Noise - 045R	1456	1458

Source 4 kHz	Source - Broad		
Noise 6 kHz w/ Source 4 kHz	Noise - 045R Source - Broad	1459	1501
Noise 4 kHz w/ Source 4 kHz	Noise - 060R Source - Broad	1503	1505
Noise 6 kHz w/ Source 4 kHz	Noise - 060R Source - Broad	1505	1507
Noise 6 kHz w/ Source 4 kHz	Noise - Broad Source - Broad	1508	1510

Element	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**
Location (inches)	0	7.5	13	22	29.5	37	44.25	51.5

Array Data Collection: Random Array

Source	**Position**	**Start Time**	**End Time**
4 kHz	Broadside	1517	1519
4 kHz	045 Relative	1520	1522
4 kHz	Fwd Endfire	1523	1525

Element	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**
Location (inches)	0	15	18.5	22	33.25	37	40	51.5

Part 3: Matched filtering

Download 6456_lab2.zip from stellar and unzip the folder. Inside, you will find two .mat files and a code template. This includes a “mystery tone” that you will need to identify, and that you will use a matched filter to characterize within the overall signal. This data is “bistatic”: the source and receiver are separated by some distance, and the returns observed in the data correspond to:

1) Direct path from the source to the receivers
2) Multipath
3) Scattering from the environment

Use the code template to write a matched filter with each of the three replicas. Add whatever code you want to plot different signal characteristics that will help you (e.g. use spectrograms and FFTs to help answer the question of which replica).

1. Matched Filter One
2. Matched Filter Two
3. Matched Filter Three